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VIDEO QUALITY OF SERVICE (QOS) USING IDEA CRYPTOGRAPHY

***[1]Hind Abdallah Osman, [2]Khalid Hamid Bilal Abdalla and [3]Afaf Mustafa Elhassan**

[1]Faculty of Science and Technology, Omdurman Islamic University, Sudan.

[2,3]Department of Communication Engineering Systems, Faculty of Engineering, University of Science and Technology, Sudan.

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***Corresponding Author**
Hind Abdallah Osman
Faculty of Science and Technology, Omdurman Islamic University, Sudan.

ABSTRACT

To maintain the stream's real-time character, a real-time multimedia transmission requires QoS guarantees. In today's IP networks, there are numerous ways for QoS provisioning. In this research, IDEL cryptography is used to improve video quality of service (QOS). Our approach addresses the issue of real-time service quality. We used an emulation system that included a streaming server, a network emulator, and video receivers to test the suggested method. The opnet simulator (in emulation mode) was utilized as a network emulator. The opnet program was used to create the streaming server and the video receiver. The results of the experiment in the test environment show that the suggested method provides sufficient QoS for HD video as well as a stable network. We discovered that IDEA provides the best user level QoS and is successful in networks with short inter–node lengths based on the assessment results.

KEYWORDS: QoS, IDEA, Video Quality Streaming, Video File Formats, Video Resolution.

I. INTRODUCTION

Many video file formats (codecs) and encryption have been created by various organizations to provide improved visual quality with small file sizes, saving storage space and allowing streaming over a slow network without buffering or delay. Users find social clouds appealing since they can share and view their personal and other information films with others in the community. The user has a variety of gadgets, including cellphones, DSLR cameras, and movie recorders, that he or she can use to capture adventure and family party movies and publish them on social media.[1] Heterogeneous devices record videos in a variety of file formats, which are then uploaded to social media via mobile apps from social media service providers.[2, 3]

Different standard video file formats codecs play an important role in providing video streaming at low bit and frame rates while maintaining high-quality services, allowing networks to be optimized, file sizes to be reduced, and compression parameters to be reduced, resulting in improved services and video quality.[4] Compression is generally based on the loss technique, which means that the compressed video loses some of the information that was present in the original video. In terms of video streaming, a video codec such as H.264 or VP8 is used to compress the video stream. Encoded video streams are collected in the container of a video file format's bit stream, such as MP4, FLV, WEBM, and others. Multiple codecs in the same file can be used to implement a range of video compression formats; many video codecs use common to make them compatible and standard video compression formats.[5] In its simplest form, digitized audio is a collection of audio samples, each of which describes the sound pressure at the time. With only 8000 samples per second[6] conversational audio can be collected and recreated with excellent precision. This means that audio can be correctly reproduced at the other end as long as the network can deliver the samples without severe jitter or packet loss. Video, on the other hand, has a significantly more complicated presentation, processing, and transport. The quality of the video is determined by factors such as brightness, contrast, color saturation, responsiveness (to motion), and lip-sync.Video samples typically take up a lot more room. Video, predictably, places a greater demand on network capacity and the transport network. The following factors influence audio quality: Microphone The headset's speaker The quality of a video call across a transport network is affected by: Camera Device for displaying information Codec for video Network of transportation Compatibility/Interoperability.

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QoS is a huge and complex topic that requires consideration of total traffic requirements (rather than simply the traffic you want to improve the quality of) and must be evaluated on every network component along the media flow's path.A video conference's visual quality is considerably more difficult to achieve since, in addition to network components, it necessitates a study and inspection of endpoint configuration and optimization. In general, video quality comprises the following.

End-point optimization- optimizing endpoint settings (e.g. resolution, frame-per-second) Carry optimization- optimizing the network to transport video traffic per network Considerations for interoperability- Endpoints with varying capabilities are frequently used in video conferences. Video quality can be influenced by how systems are designed and configured to maximize interoperability The QoS considerations on the gateway when handling video calls will be the subject of this publication.[8]

Endpoint tuning entails adjusting a set of parameters on the video endpoints.

Objectivity (i.e. picture size)

Frame rate (also known as motion sensitivity or reality)

The act of tagging (i.e. ToS marking)

Tuning a network for video usually entails the following steps:

Understanding the traffic flow through the - for example, peak [call] volume, etc.

II. IDEA

The International Data Encryption Algorithm (IDEA) was designed to replace the Data Encryption Standard (DES) (DES). IDEA is a minor improvement to the Proposed Encryption Standard that was previously known as the Improved Proposed Encryption Standard (IPES) (PES).

For encryption and decryption, IDEA employs identical methods, with the exception of certain inverted round key ordering. It has an 8-round format and uses a 128-bit key to operate on 64-bit blocks. Until the key schedule was updated, IDEA had weak keys, and it may require further revision in the future.

Pretty Good Privacy has been and continues to be compatible with IDEA (PGP). The IDEA NXT algorithm, which was previously known as FOX, has supplanted IDEA.

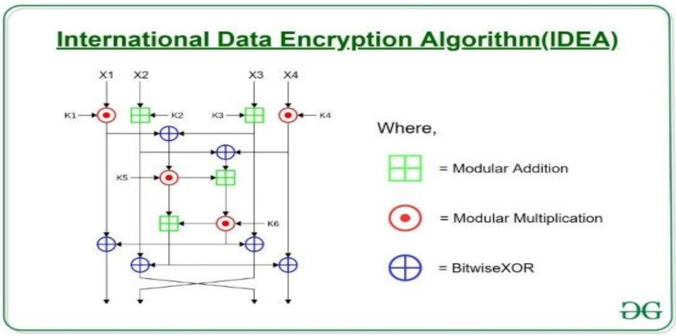

Figure 1: IDEA Algorithm

III. Literature Review

1- An encryption technique for video compressed using H.264 was proposed in this research to securely exchange highly secret video. In order to demonstrate the usefulness of the suggested algorithm, a comparison analysis of the new algorithm with or existing algorithms has been presented in this work.[9]

2- This paper will describe voice over internet protocol to a level that allows discussion of security issues and business concerns of implementing VoIP components of a VoIP system, as well as relevant security issues and concerns as they apply to the topics explored call process are gate ways and two of the more common architecture.[10]

3- This paper shows how to create a video over IP application with a Compaq IPAG and a field programmable gate array for data processing. The video data is transmitted directly from the camera to the network card via the gate array, obviating the need for the CPU to do extra computations that normal video over IP communication software imposes.[11]

4- This paper will describe the voice over internet protocol (VoIP) in order to demonstrate how a VoIP system might benefit a business. The issues that will be of concern to the business will be those that affect service quality. Equipment and VoIP components will be included. A quick overview of VoIP technology, including network components, call processors, gateways, and two of the most prevalent topologies.[12]

5- The investigation of real-time transmission and fruition methods of speech contents across an internet protocol is the focus of this paper.[13]

6- The purpose of this thesis is to examine the quality of service performance and its impacts when video is transmitted via GBR (guaranteed bit rate) and multiple GBR bearers over LTE using OPNET (Optimized Network Engineering Tool).[14]

7- The work in this thesis focuses on techniques for improving various features of the dynamic adaptive video streaming over HTTP (DASH) service as outlined by research topics connected to QOS. The high QOS needs of video services are the driving force behind focusing on them.[15]

8- This research investigates the use of encryption algorithms at various levels of compression. This real-time video compression and encryption provides security for applications such as video conferencing, surveillance camera data protection, and so on. The examination of adding numerous encryption algorithms in the compression phases, such as transformation and coding stages, was provided using the generic structure of a video encoder.[16]

9- The authors of this paper propose a solution for supporting the broadcast of multiple video channels that can only be accessed by authorized users; additionally, when a video channel is not visualized in the last mile, it may be temporarily disabled so that the cable can be used for other services, such as standard internet access.[17]

10- The available video information protection technologies are examined in this research. The current H.323 protocol stack has been examined. It has been suggested that a new encryption/decryption layer be implemented. A partial data encryption technique for the H.323 protocol stack is offered, as well as an example architecture for a Virtual Private Video Network.[18]

11- **SUMMARY**

In a traditional study, the video was compressed using any compressed application, such as H.264. Compaq IP age is being used for data processing. Regular video over IP connection software imposes extra delay and computation requirements on the CPU, therefore video data is transmitted directly from the camera to the network card via the gate array. OPNET (Optimized Network Engineering Tool) was used to examine the quality of service utilizing GBR (Guaranteed bit rate) and additional GBR bearers over LTE, or an analysis of the current H.323 protocol stack was performed.

IV. Conceptualization (Methodology)

We conducted numerous experiments utilizing various movies on social clouds to determine the QoS of video streaming. Technical work based on HD videos in three resolutions and quality levels: 360P, 720P, and 1080P.[19]

Table 1: QoS of video streaming			
Video name	**Resolution**	**Frames Per Second(fps)**	**Data transfer speeds and duration (MB/s)**
Bronz	1920 x 1080	60	21,67
Gold	1600 x 720	70	19,75
silver	1280 x 360	75	15,45

V. RESULT AND DISCUSSION

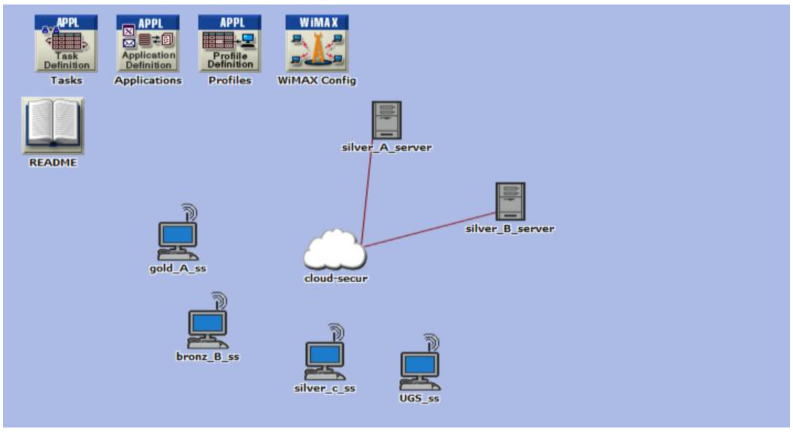

Figure 2: WiMAX architecture.

Table2: Video Conferencing. Packet Delay Variation		
statistic	With_IDEA: Video Conferencing. Packet Delay Variation	Witeou_IDEA: Video Conferencing. Packet Delay Variation
Video Conferencing .Packet Delay Variation	0.000945457	35.76862271

VI. ANALYSIS RESULT

Analysis of the video's impact due to IDEA coding On the simulated environment opnet, we assess and analyze the impact of the IDEA interaction among connections with different QoS service classes, and optimize secure video coding. The measurements are taken when IDEA is active, and they are repeated while the network is protected by the IDEA secure algorithm.

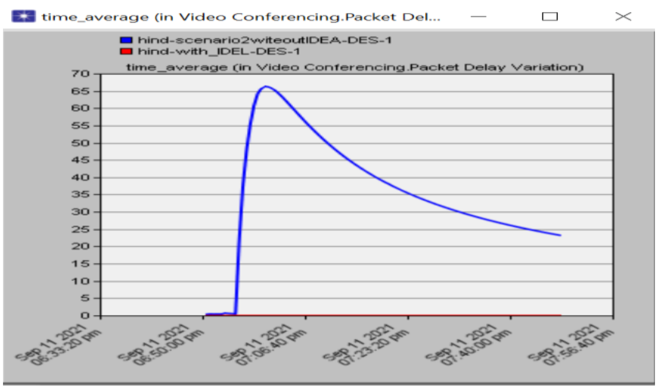

Figure 3: Video Conferencing. Packet Delay Variation.

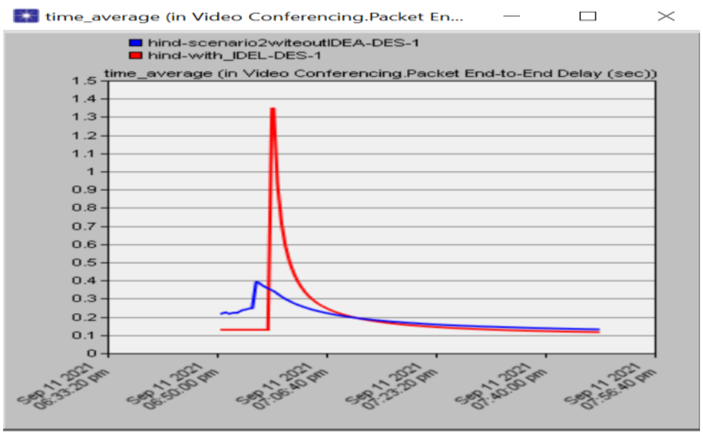

Figure 4: Video Conferencing. Packet End-to-End Delay (sec).

Table 3: Video Conferencing. Packet End-to-End Delay (sec)		
Statistic	With_IDEA: Video Conferencing .Packet End-to-End Delay (sec)	Witeout_IDEA: Video Conferencing. Packet End-to-End Delay (sec)
Video Conferencing .Packet End-to-End Delay (sec)	0.196509111	0.190186461

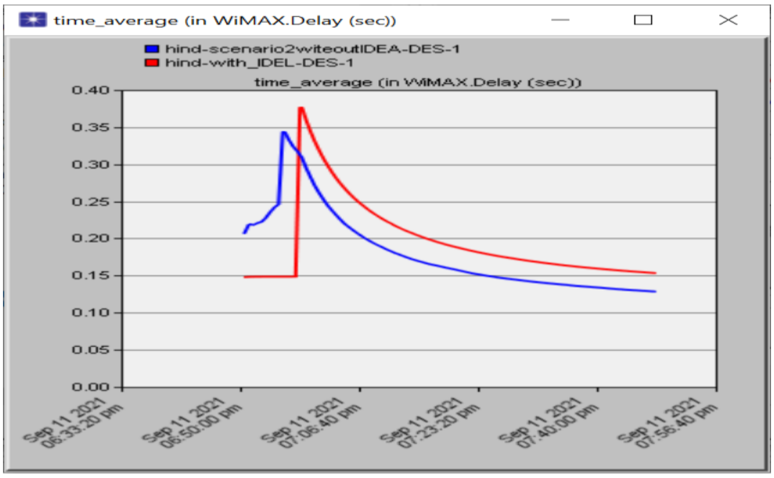

Figure 5:WiMAX.Delay (sec)

Table 4: WiMAX.Delay (sec)		
	With_IDEA	**Witeout_IDEA**
Statistic	0.194624703709	0.180115456816

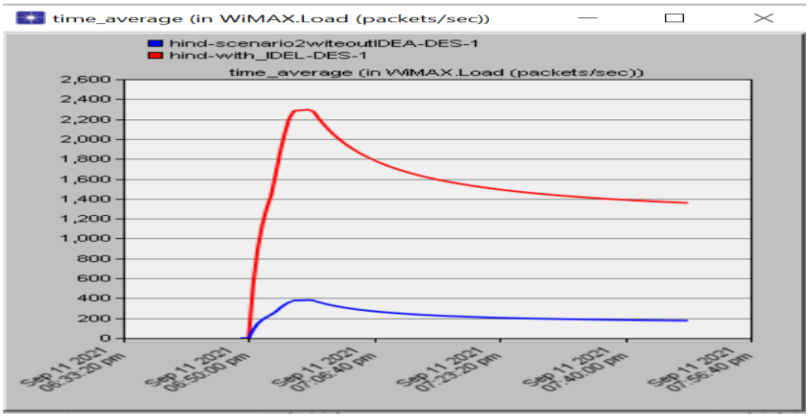

Figure 6: WiMAX.Load (packets/sec).

Table 5: WiMAX.Load (packets/sec).		
	With_IDEL	**Witeout_IDEA**
Statistic	1,545.53418974	223.6334823

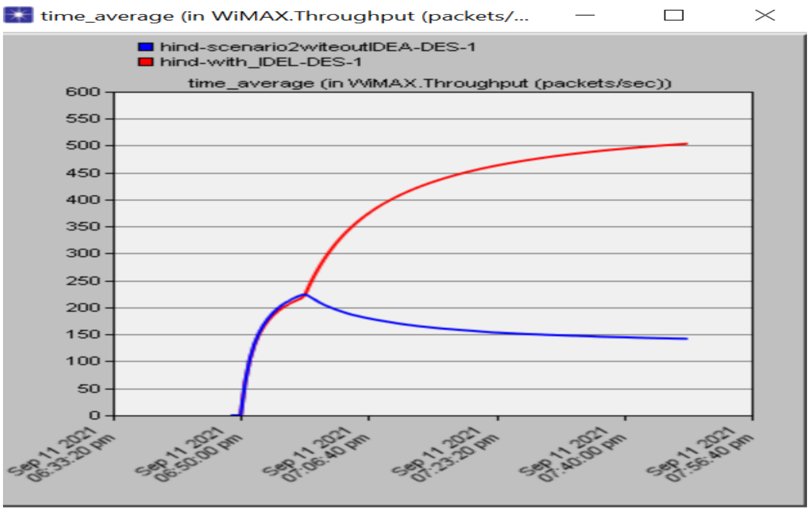

Figure 7: WiMAX.Throughput (packets/sec).

Table 6: WiMAX.Throughput (packets/sec).		
	With_IDEA	**Witeout_IDEA**
Statistic	389.871430872	157.985869225

VII. QOS Analysis

The following are the recommended network SLAs for video quality on transport networks.[20]

- Latency ≤ 150–300ms
- Jitter ≤ 10 ms–50ms
- packet loss ratio ≤ 0.5%
- Packet loss ratio= Number of lost packets / Number of received packets

The following are the network SLAs for video quality on transport networks that are recommended.

Teble 7: Speed Test Results of Evolved IDEL Network of server A and B.

No.	Time	Provider A	Provider B
		Download	Download
1	60.00	5,532 Mbps	7,620 Mbps
2	05.00	208 Mbps	9,332 Mbps
3	20.00	3,244 Mbps	4,958 Mbps
4	30.00	3,848 Mbps	5.104 Mbps

Using a range of customers and providers to test throughput.

Teble 8: Average throughput from client to UGS Server A and B.

No.	Name of Client	server A	server B
		Average Throughput (Kbps)	Average Throughput (Kbps)
1	Bronz	446.354	444.417
2	Gold	424.979	441.052
3	silver	405.319	407.069

Delay in testing due to a wide range of clients and servers Simultaneously with the sampling of data throughput, jitter, and packet loss, scenarios of delay testing based on various client and server kinds are carried out. Table 9 and Table 10 exhibit the test data delay variety based client and server that have been averaged.

Table 9: Average delay based on variety of clients server A

No	Name of Client	Average Delay End to End (ms)
1	Bronz	31,153
2	Gold	32,949
3	silver	32,913

Table 10: Average delay based on variety of clients server b.

No	Name of Client	Average Delay End to End (ms)
1	Bronz	28,317
2	Gold	28,895
3	silver	32,913

Jitter is being tested on a variety of clients and servers. Simultaneously with the sampling data of throughput, latency, and packet loss, a Jitter test scenario based on various client and server configurations is carried out. Table 11 and Table 12 exhibit the averaged data of jitter test results from a variety of clients and providers.

Table 11: Average jitter from Client to UGS Server A dan B.			
No.	Name of Client	server A	server B
		Average Jitter(ms)	Average Jitter(ms)
1	Bronz	22.112	17.782
2	Gold	22.458	17.762
3	silver	21.345	17.639

Table 12: Average jitter from UGS Server to Client Provider A dan B			
No.	Name of Client	server A	server B
		Average Jitter(ms)	Average Jitter(ms)
1	Bronz	29.725	30.935
2	Gold	32.879	31.338
3	silver	33.954	33.024

A number of clients and servers were used to test packet loss.

Table 13 and Table 14 illustrate the test data packet loss based on client and provider variety, which is conducted concurrently with the sampling of data throughput, delay, and jitter that has been averaged.

Table 13: Average Packet Loss from Client to UGS Server A dan B			
No.	Name of Client	server A	server B
		Average Packet Loss	Average Packet Loss
1	Bronz	0.282%	0.241%
2	Gold	0.257%	0.184%
3	silver	0.230%	0.263%

Table 14:Average Packet Loss from UGS Server A dan B			
No.	Name of Client	server A	server B
		Average Packet Loss	Average Packet Loss
1	Bronz	1.080%	0.062%
2	Gold	1.019%	0.720%
3	silver	0.578%	0.541%

VIII. CONCLUSION

The quality of network services utilized for live video streaming is substantially affected by the test time. Based on the number of clients, it is established that the higher the number of clients, the lower the throughput, with the highest throughput of 467.750 Kbps when accessed by one client using provider A. Based on the number of clients, throughput from client to UGS Server appears to be unstable, with the highest throughput value of 483 000 Kbps when accessed by three clients utilizing server B. The End-to-End Delay is determined by the number of customers accessing the live video streaming; the bigger the number of clients accessing the live video streaming, the longer the delay. The End-to-End Delay is determined by the number of customers accessing the live video streaming; the bigger the number of clients accessing the live video streaming, the longer the delay. When reached by one client using provider B, the lowest value of End to End Delay is 27.389 ms, while the greatest value of Delay End to End is 36.180 2 ms when accessed by two clients using server A. The Jitter from Media Server to Client is depending on the number of clients, and it is discovered that the more the number of clients, the higher the jitter, with the lowest jitter of 27.339 ms when accessed by one client using server A. Based on the grouping of the delay, the jitter and the packet loss from Telecommunications and Internet Protocol Harmonization Over Networks (WIMAX) standard, the Video Quality of Service (QOS) using IDEA cryptography on both of the providers used for live video streaming is categorized good.

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